DIGITAL AUDIO PROCESSING FLOW FOR THE 560-T

The following block diagram illustrates the audio processing flow for the 560-T transmitter.

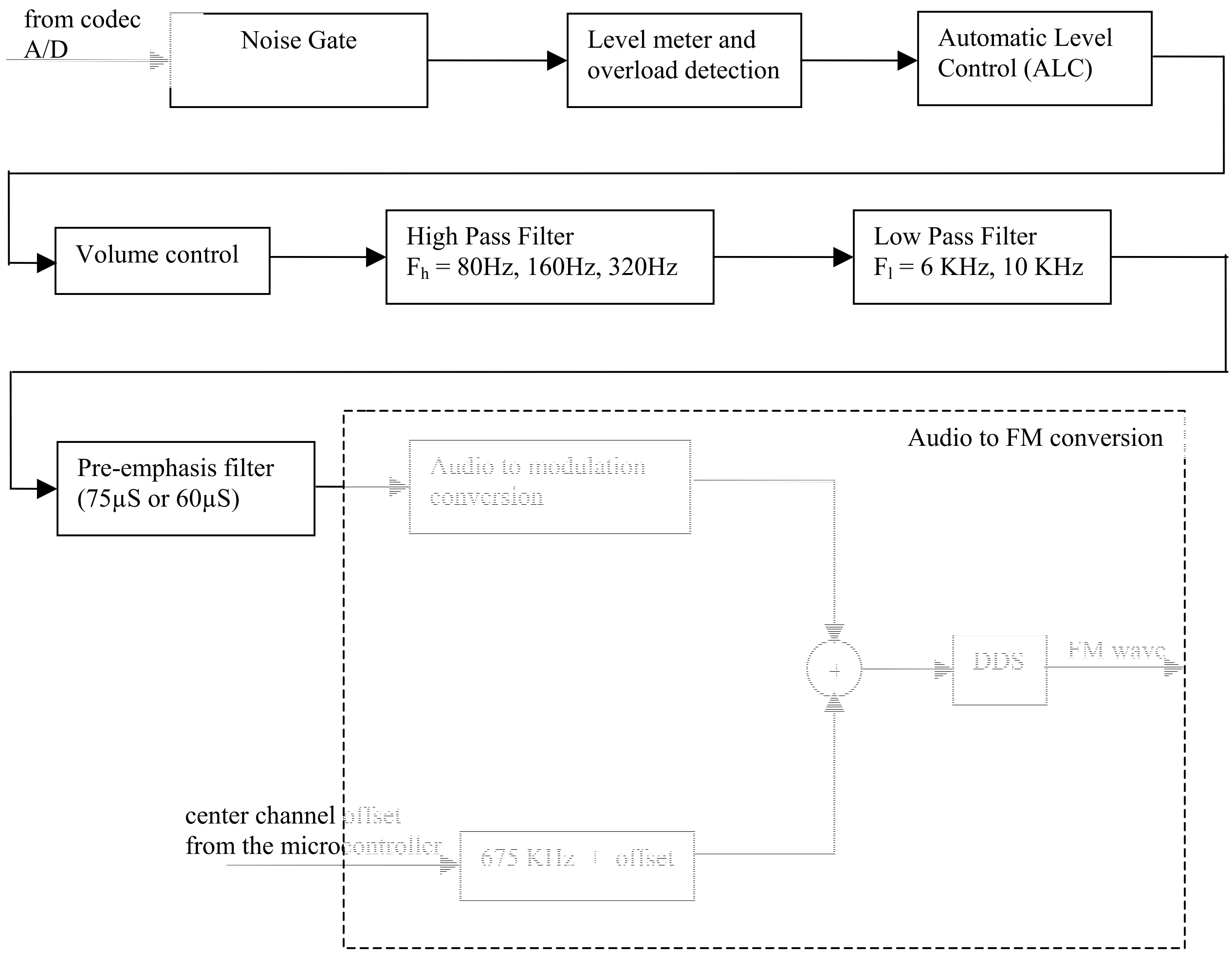

Figure-1: Audio processing flow for the 560-T transmitter.

Noise Gate: The incoming audio signal level is compared with a preset threshold. If the incoming signal level is less than the threshold the noise gate is switched on, which will make the output to be all zeros. If the incoming signal is higher than the threshold, the audio data is forwarded to the next processing block. The noise gate's attack time is 6.5ms, and it has an immediate release. The noise gate effectively removes any spurious modulation caused when the system has no valid input (just noise) and the gain of the system is set high.

Level meter/overload detection: The audio level of the input signal is calculated on a log scale. The microcontroller reads this value and displays it as a bar graph on the front panel. Also when the input produces more deviation than allowed deviation (peak deviation is 5 KHz for narrow band and 20 KHz for wide band), a red LED on the front panel is illuminated which indicates the overload condition. If the overload indicator is active for more than 3 seconds, the DSP will digitally reduce the FM deviation by 50%. This way, the output spectrum will never spill into neighboring bands. When the overload condition no longer exists the device will automatically resume its normal operation.

Automatic Level Control (ALC): The ALC algorithm's main objective is to prevent digital clipping. Digital clipping will cause undesired harmonics. ALC algorithm makes sure that the current gain level will never cause the current input to clip inside the DSP. If necessary the ALC will reduce the gain temporarily. ALC operated only when the gain is set to some positive value.

Volume control: Volume control applies the user set gain to the input signal. The user set gain is ignored if the ALC is in effect; instead the ALC computed gain is used.

High Pass Filter (HPF): The digital HPF has user selectable cutoffs of 80 Hz, 160 Hz, and 320 Hz. The analog audio processing front end provides for the 40 Hz high pass filter. The purpose of the HPF is to attenuate low frequency noise sources like fan noise, air conditioner noise, 60 Hz hum, etc.

Low Pass Filter (LPF): The digital LPF has user selectable cutoffs of 6 kHz, or 10 kHz. The LPF provides modulation limiting by eliminating all high frequency signals that may cause over modulation.

Pre-emphasis filter: Pre-emphasis filters provide a pre-emphasis of 75µS in narrow band mode and 60µS in the wide mode. The non standard value of 60µS is chosen for the wide band mode to make sure that the transmitter works best with Phonic Ear receivers (optimum frequency response for the overall system).

Audio to FM conversion: The microcontroller provides the center channel offset frequency. This offset is added to the base frequency of 675 KHz, to get the new output channel frequency. The filtered audio data is multiplied by a pre-computed modulation constant, to achieve the frequency deviation. This deviation is then added to the output channel frequency to arrive at the final output frequency, and the composite data is sent to the DDS. The DDS outputs the FM waveform.

Frequency response curves:

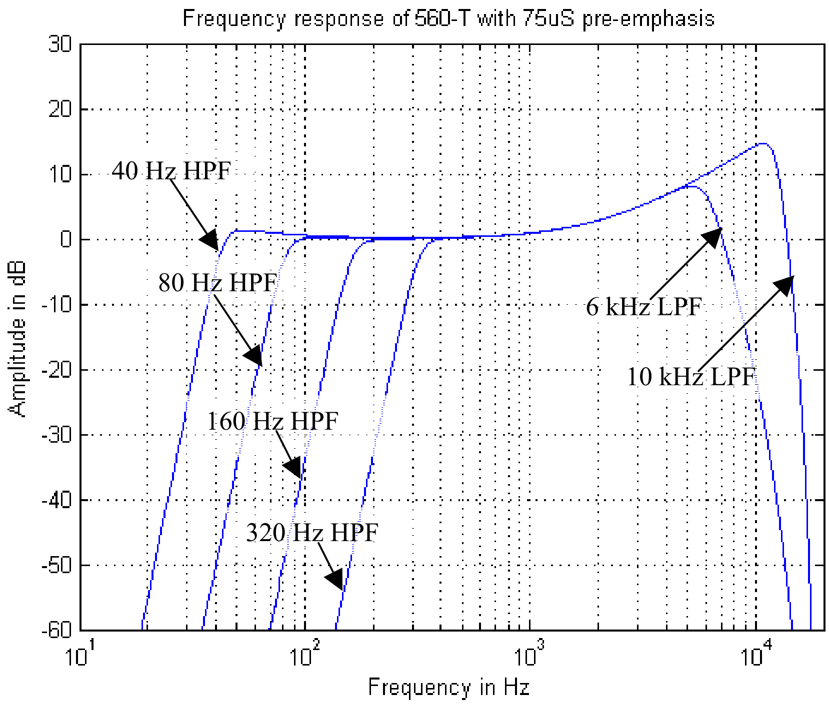

Figure-2: Frequency response of 560-T with 75μS pre-emphasis (narrow band).

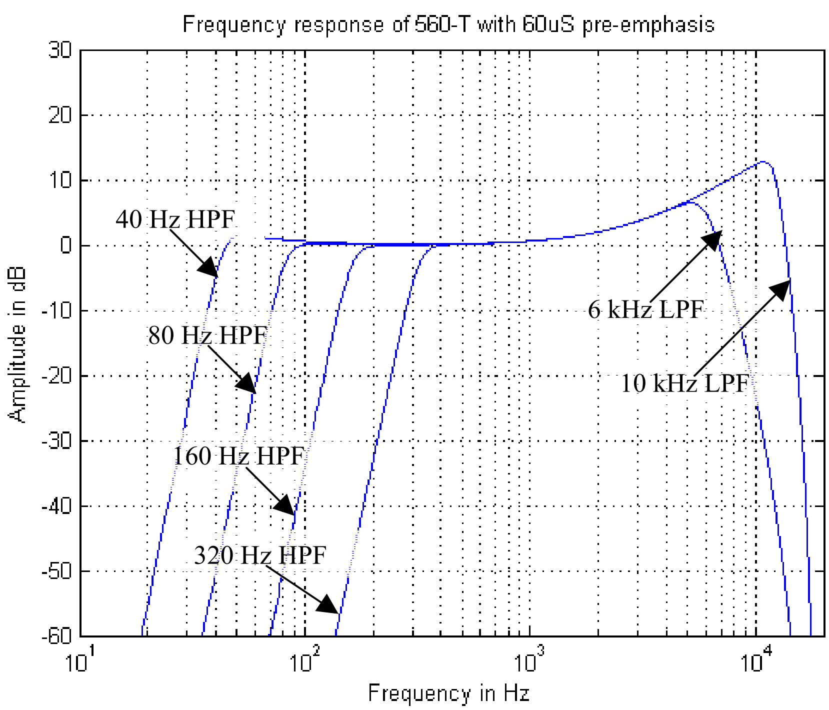

Figure-3: Frequency response of 560-T with 60μS pre-emphasis (wide band).